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# UNITED STATES PATENT APPLICATION

for

## A METHOD AND APPARATUS FOR DYNAMIC DSP RESOURCE MANAGEMENT

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BACKGROUND OF THE INVENTION

**Field of the Invention**

This invention relates to resource allocation within a computer or other network. More particularly, the present invention relates to the allocation of DSP Resources in a Digital Communications Network.

**Description of the Related Art**

Figure **1** generally depicts a digital network adapted to transport voice, data and other information. The network includes a plurality of Customer Premises Equipment nodes **100, 108** (hereinafter "CPE nodes"), framer modules **101, 107**, Network Interface Systems **102, 106** (hereinafter "NISs"), DPS resources **103, 105** within each of the NISs, and network **104**.

When a communication connection is requested from CPE node **100** to CPE node **108** the signal first travels from the CPE node **100** to framer module **101** over digital T1 carriers **110, 120**. Before being transmitted over T1 carriers **110, 120**, the signal from CPE node **108** is sampled and converted to a digital signal. A common sampling rate to convert the analog signal to digital is 8000 samples per second, with each digital sample represented by 8 bits of data. Thus, the data rate of the new digital signal is: 8000 samples/sec x 8 bits = 64,000 bits/sec. This technique is known as Pulse Code Modulation (hereinafter

"PCM") and is used extensively throughout the backbone of the modern telephone system.

Although no international standard has been adopted, the T1 carrier is one method of PCM used throughout North America and Japan. The T1 carrier is comprised of 24 channels of digital data multiplexed together. As shown in **Figure 2**, digitally sampled data from each of the 24 channels are packaged into successive frames of 8 bits/channel x 24 channels + an additional framing bit **210** = 193 bits. Outside of North America and Japan a similar standard known as E1, is commonly implemented. E1 operates in a manner similar to T1 except that it uses 32 8-bit data samples (i.e., 32 channels) instead of 24.

After the signal has been sampled, converted to a digital signal and transmitted over a T1 carrier **110**, as described above, it is then transferred to a framer module **101**. The framer module **101** serializes the digital data from the plurality of T1 carriers **110** into a bit stream which is sent to NIS **102**. An example of a known framer module is a Rockwell™ BT8370. In the NIS **102** the digital data samples are packaged according to the protocol used in network **104** (e.g., ATM). Additionally, although PCM by itself provides for a data compression rate of 8:1, it is often desirable to further compress the digital PCM data at NIS **102** in order to save bandwidth over the network. This can be accomplished using Digital Signal Processing (hereinafter "DSP") resource blocks **103, 105** within NIS **102**. For example, if the 64,000 bits/second PCM

signal is further compressed by DSP resource **103** at a compression ratio of 16:1 the resulting digital signal will be transmitted at 4,000 bits/second. This represents a significant reduction in required bandwidth across network **104** to transmit the same underlying signal. Such compression techniques are particularly useful in networks which are heavily loaded with network traffic. Examples of compression algorithms known in the art include the International Telegraph Union (hereinafter "ITU') standards G.711, G.726, G.729-A, G.729, and G.728.

However, there is a tradeoff between bandwidth savings over network **104** and the implementation of costly DSP resources **103** in NIS **102**. In general, the higher the compression ratio required by the compression algorithm, the more DSP resources **103** are used up processing the compression request over a given period of time. Thus, referring to the table in **Figure 3**, while a single DSP resource can process 16 channels of data if no compression is used (i.e., in baseline PCM mode), it can only process 5 channels if data is compressed at 2:1, and only 2 channels if the PCM signal is compressed at 8:1.

This is because the DSP resource has a limited amount of time in which to compress the data within a T1 frame before it must move on to the next frame of data. Referring to **Figure 4**, 'T' represents the maximum amount of time that a DSP resource may spend compressing data from a frame before it must move on to the next successive frame (i.e., the data from frame 1 must be compressed

before frame 2 arrives). While a single DSP resource can process 16 T1 channels per unit of time 'T' in baseline PCM mode (i.e., each channel requiring approximately 'T'/16 to process), the same DSP resource can only process one T1 channel per the same unit of time 'T' running at a compression ratio of 16:1. Thus, the chosen compression ratio will have a significant impact on DSP resource usage.

Following compression, the data samples are delivered through network **104** to NIS **106**, where the data is decompressed and passed on to framer module **107**. Framer module **107** reconstructs the original T1 signal which it transfers to CPE **108** over a T1 carrier **120**. The system is bi-directional to ensure 2-way communication.

The management of the DSP resources in prior systems occurs without regard to whether the DSP resources are depleted. In other words, when multiple calls are received from the CPEs **100**, the NIS **102** may have its DSP resources **103** consumed quickly. Once all DSP resources **103** in NIS **102** are allocated, any attempt to open a new communication channel through NIS **102** will result in a busy signal. What is needed is a DSP resource allocation system to ensure that there are always enough DSP resources **103** available in NIS **102** to process the incoming data by reserving enough DSP resources to process all idle T1 channels in PCM mode. In sum, a scheme for managing the DSP resources

**103** in NIS **402** in order to reduce the possibility that a call request will be rejected is needed.

## SUMMARY OF THE INVENTION

The present invention relates to a system and method of managing DSP resources within a network interface system (NIS). A Digital Signal Processing (DSP) resource manager dynamically allocates DSP resources to ensure that a DSP resource is always available to process an incoming call. The DSP resource manager reserves a sufficient number of DSP resources in a PCM group in order to ensure processing of any active communication channels. Each time a call is received and assigned to a T1 channel, the DSP resource manager recalculates the number of idle channels and the number of PCM resource channels which must be reserved to ensure processing of all idle channels. The DSP resource manager also recalculates DSP resource availability in the PCM group each time a call is deactivated. By tracking the number of idle channels in the system as well as the number of idle DSP resource channels in the PCM group, the DSP resource manager can reallocate the resources as necessary to optimize the ability of the system to fulfill enhanced compression mode requests, while also ensuring that there are sufficient resources available to process any call in PCM mode.

## BRIEF DESCRIPTION OF THE DRAWINGS

A better understanding of the present invention can be obtained from the following detailed description in conjunction with the following drawings, in which:

FIG. **1** shows generally a data network.

FIG. **2** shows a T1 carrier signal.

FIG. **3** shows a table illustrating the relationship between data compression ratio and channels processed per DSP resource.

FIG. **4** shows a graph illustrating the relationship between data compression ratio and time required to compress data by DSP resources.

FIG. **5** illustrates a T1 carrier and a network interface system.

FIG. **6** is a flow diagram illustrating DSP resource manager logic when a new call is received.

FIG. **7** is a flow diagram illustrating DSP resource manager logic when a call is deactivated.

## DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

A system and method of managing the DSP Resources in a Network Interface System (NIS) is described. Referring to **Figure 5**, a DSP resource manager **500** is used to dynamically allocate DSP resources **501** from a DSP resource block **502** to ensure that a DSP resource **501** is always available to process an incoming call.

As a call request is received by NIS **102**, DSP Resource Manger **500** attempts assigns the call to DSP resource block **502** for data compression. In one embodiment, incoming calls are delivered through channelized T1 carriers **504**. There may be eight T1 carriers **504**, with each T1 carrier channelized into 24 channels **506**, and a resource block **502** comprised of 26 DSP resources **501** (i.e., chips). However, one of ordinary skill in the art will readily recognize from the following discussion that, depending on the system configuration, a different number of carriers, channels or DSP resources could be employed without departing from the underlying principles of the present invention. Moreover, while the embodiment discussed below uses a T1-based transmission scheme, other transmission schemes such as E1, T3, etc., could be implemented as well. Using 8 T1 carriers **504**, the present configuration provides for 8 carriers X 24 channels = 192 total channels to be managed by the DSP resource manager **500**.

In the present embodiment, the DSP resource manager **500** allocates the 26 DSP resources **501** in such a way as to ensure DSP resource availability to process

any incoming call in PCM mode. The 26 DSP resources **501** from resource block **502** are divided into two groups: a PCM group **510** and a Compression Group **512**. The PCM Group **510** is a group of DSP resources that have been reserved to process incoming T1 channels in baseline PCM mode (non-compression mode). Thus, the PCM group **510** is made up of some variable number of the 26 DSP resources **501**. Resources **501** are allocated to the compression group **512** to process calls requesting compression ratios other than baseline PCM. Accordingly, if N of the DSP resources **501** are assigned to the PCM group **510**, then the compression group **512** is comprised of 26-N of the DSP resources **501**.

The number of DSP resources **501** allocated to the PCM group **510** will depend on the number of T1 channels that are active (ready to receive incoming/outgoing calls) and the number of active channels requesting compression. To illustrate the operation of the DSP resource manager **500**, if it is initially assumed that all of the 192 T1 channels are active and that each DSP resource **501** can process 16 channels in PCM mode, and that no channels are currently allocated to the compression group, a total of 192/16 = 12 DSP resources **501** must be allocated to the PCM group **510** to ensure that any call requesting PCM mode can be processed. This means that 26 - 12 = 14 DSP resources **501** are initially allocated to the compression group **512**. Accordingly, if the first 28 incoming calls request 8,000 bit compression (8:1), then all of the 14 DSP resources **501** in the compression group **512** will be used up. This is because

at 8,000 bit compression, each of the DPS resources can only process 2 T1 channels (see **Figure 3**).

If a 29th call received by the NIS **102** now requests compression, the call will be rejected in prior art systems because all of the DSP resources **501** in the compression group **512** are busy. However, the present DSP resource manager **500** will attempt to reallocate DSP resources **501** within the resource block **502**. Specifically, DSP resource manager **500** will recalculate the number of DSP resources **501** required in the PCM group **510** based on the number of active T1 channels and the number of active channels assigned to the compression group. Assuming that 28 incoming calls have been allocated to the compression group as stated above, then 192 (total active channels) – 28 (active calls assigned to compression group) = 164 active channels which must be provided for by the PCM group **510**. Since each DSP resource **501** can process 16 calls in PCM mode, 164/16 = 10.25 DSP resources **501** are required if all 164 channels transmit calls requesting PCM mode. This value is rounded up to 11 because a single DSP resource cannot be split between the compression group **512** and the PCM group **510**. Accordingly, once DSP resource manager **500** reevaluates the DSP resource allocation scheme it will reassign one of the 12 DSP resources **501** in the PCM group **510** to the compression group **512** and call 29 will be processed using the compression algorithm which it requested. By continually reevaluating DSP resource allocation in this manner, the DSP resource manager **500** provides

flexibility for processing calls and at the same time ensures that there are always sufficient DSP resources **501** to process any new call in PCM mode.

Assuming that DSP resource **501** has been reallocated from the PCM group **510** to the compression group **512** as described above and is operating at full capacity (e.g., processing calls 29 and 30 using 8,000 bit compression), when a new call request (i.e., call 31) is sent from a CPE node **100** requesting compression, DSP resource manager **500** will not be able to reallocate another DSP resource **501** to handle the compression request. This is because [192 (active channels) – 30 (channels using compression group)]/16 (channels per DSP resource **501**) = 10.125. Thus, all 11 DSP resources **501** currently allocated to the PCM group **510** are required to stay there to ensure PCM availability. Accordingly, DSP resource manager **500** will send a response to CPE node **100** indicating that no compression resources are available. However, the response may also indicate that a DSP resource **501** from the PCM group **501** is available. At this point CPE node **100** can choose to establish communication using the DSP resource **501** in the PCM group **510** or can try back at a later time (when a DSP resource **501** from the compression group **512** may be available). In this way, DSP resource manager **500** provides a caller with an option rather than simply denying the call request outright (e.g., with a busy signal).

Up until now it has been assumed that all 192 T1 channels **506** (provided by the 8 T1 carriers **504**) are active. In one embodiment, however, DSP resource

manager **500** is capable of activating and deactivating one or more T1 channels **506** based on network traffic requirements. Significantly, when the DSP resource manager **500** reserves enough DSP resources **501** in the PCM group **510** as described above, it is doing so based on the number of *idle* channels of *active* T1 carriers rather than *inactive* channels of *inactive* T1 carriers. In other words, an idle channel is a channel on an active T1 line which is not currently in use (i.e., not currently supporting a call). By contrast, an inactive channel is a channel on an inactive T1 line. Inactive channels are not factored into the equation used to reserve DSP resources **501** in the PCM group **510** until a request to activate a new T1 line is received. When this happens, additional DSP resources **501** must be allocated for the additional channels. With these new variables in mind, the following is a detailed description of the logic followed by the DSP resource manager upon receiving a new call request and a call deactivation request.

**Figure 6** is a flowchart which illustrates the steps taken by DSP resource manager **500** when a new call request is received. When a new call is received at step **602**, DSP resource manager **500** first determines whether or not there is an open channel on an active T1 line at step **604**. If there is an open channel on an active T1, DSP resource manager **500** checks the compression rate requested at step **610**. If there is no channel available on an active T1 line, DSP resource manager **500** attempts to initialize a new T1 line. First, DSP resource manager **500** determines if there are sufficient DSP resources **501** available to process all channels if a new T1 line is activated **605**. If not, the caller is notified that there

are insufficient resources available and the call is not activated (step **606**). If there are sufficient DSP Resources **501** available to process all channels of a new T1, DSP resource manager **500** activates the T1 line at step **607**. Each time a new T1 line is activated, DSP resource manager **500** allocates sufficient DSP resources **501** to the PCM group **510** to ensure that if all channels become active (i.e., receive calls), the PCM group **510** will have sufficient DSP resources **501** to process the calls (step **608**).

As previously stated, a single DSP resource **501** is capable of processing 16 channels in PCM mode. Accordingly, for a T1 carrier **504** which is channelized into 24 channels **506**, two DSP Resources must initially be reserved in the PCM group **510** to ensure processing of the newly activated T1 carrier at step **608**. However, if a second T1 line is activated, only one additional DSP resource **501** needs to be allocated since the processing capability in one of the two DSP resources **501** initially reserved when the first T1 line was activated is only being partially used. That is, the second DSP resource **501** is still capable of processing an additional 8 channels: 24 (channels/T1 carrier) – 16 (channels/DSP resource) = 8 (additional channels that the DSP resource can process).

At step **610** DSP resource manager **500** checks the compression rate requested by the incoming call. If PCM is requested (step **612**), the call is assigned to a channel from a DSP resource in the PCM group **510** (step **613**). DSP resource manager **500** then updates two data values which it uses to keep track

of DSP resource availability. In step **614** it decreases "the number of idle voice channels" variable by one **614**, and then in step **615** it decreases the "number of idle DSP resource channels in PCM group" by 1.

If at step **612** DSP resource manager **500** determines that the incoming call requests compression, it attempts to assign the call to the compression group **512**. In step **616**, DSP resource manager **500** determines if there are sufficient DSP resources **501** in compression group **512** to process the call at the highest specified compression rate (the call may request more than one compression rate) in order to save bandwidth across network **104**. If DSP resources **501** in compression group **512** are insufficient to implement any of the requested compression algorithms, DSP resource manager **500** notifies the CPE node **100** requesting compression. If PCM transmission through network **104** is adequate, CPE node **100** can accept a channel using PCM mode (recall that a PCM channel should always be available). The call is then assigned to PCM group **510** at step **613**. If, however, PCM is inadequate CPE node **100** rejects PCM mode and can try back at a later time (starting from step **600**). In this case, the caller may receive a busy signal.

If there are sufficient DSP resources **501** in the compression group **512** to process the request at step **616**, the call is assigned to the compression group **512** (step **618**). Once a call is assigned to the compression group **512**, DSP resource manager **500** decrements the "number of idle voice channels" variable by one

(step **620**). DSP resource manager **500** then recalculates the number of DSP resources **501** required in the PCM group **510** to process all idle voice channels. If PCM group **510** contains more resources **501** than necessary, they are reallocated to the compression group **512** in step **624**).

**Figure 7** is a flowchart which illustrates the steps taken by DSP resource manager **500** for a call deactivation. DSP resource manager **500** is notified when a call has been deactivated. If a call has been deactivated (step **710**), the "number of idle voice channels" variable is incremented by one in step **712**. If the deactivated call was handled by a DSP resource **501** in the PCM group **510** (step **714**), the "number of idle DSP resource channels available in the PCM group" variable is also incremented by one (step **716**), since the deactivation of the call has opened a DSP resource **501** channel in the PCM group **510**.

If the call was being handled by a DSP resource **501** in the compression group **512**, DSP resource manager **500** recalculates the number of DSP resources needed in the PCM Group **510** at step **720** (i.e., so that there are sufficient DSP resources **501** to process any calls received on the idle voice channels). If there are sufficient DSP resources **501**, no reallocation is necessary (step **722**). If, however, there are insufficient DSP resources **501**, DSP resource manager **500** determines whether there are any idle DSP resources in the compression group **512** (step **724**). If so, the idle DSP resources **501** are reallocated from the

compression group **512** to the PCM group **510** (step **726**). If not, DSP resource manager **500** does not reallocate (step **722**).

A scheme for DSP resource allocation has been described. As disclosed, a DSP resource manager **500** ensures that a sufficient number of channels are available for PCM processing of any channel on an active T1 line. In one embodiment, tracking and reallocation are done on a call by call basis. Reallocation operations are performed after each call is activated or deactivated. Recalculation and reallocation may also take place as each T1 line is activated, or at some other interval during processing.

One of ordinary skill in the art will readily recognize from the following discussion that alternative embodiments of the structures and methods illustrated herein may be employed without departing from the principles of the invention. Throughout this detailed description, numerous specific details are set forth such as specific digital carrier types (i.e., T1) and network protocols (i.e., ATM) in order to provide a thorough understanding of the present invention. It will be appreciated by one having ordinary skill in the art, however, that the present invention may be practiced without such specific details. In other instances, well known data communication techniques have not been described in detail in order to avoid obscuring the subject matter of the present invention. The invention should, therefore, be measured in terms of the claims which follow.